Claims

[c1] What is claimed is:

1.A digital audio signal processing method for increasing an associated processing speed, comprising:

presetting a first impulse response with regard to a frequency response of a digital audio signal, said first impulse response having a plurality of first sampling points in time domain, a total number of said first sampling points equaling a first predetermined value, each first sampling point corresponding to a first amplitude;

establishing a second impulse response by selecting a plurality of first sampling points and related first amplitudes from said first impulse response to function as second sampling points and related second amplitudes of said second impulse response, a total number of said second sampling points being less than said first predetermined value; and

processing said audio signal in time domain by said second impulse response according to a predetermined algorithm.

[c2] 2.The digital audio signal processing method of claim 1 being applied on a digital equalizer.

[c3] 3.The digital audio signal processing method of claim 1 wherein said first sampling points are spaced by a fixed interval in time domain.

[c4] 4.The digital audio signal processing method of claim 1 wherein an average power of said second amplitudes associated with said second impulse response is greater than a predetermined percentage of an average power of said first amplitudes associated with said first impulse response.

[c5] 5.The digital audio signal processing method of claim 4 wherein said predetermined percentage is 99%.

[c6] 6.The digital audio signal processing method of claim 1 wherein said digital audio signal is generated by performing a pulse code modulation (PCM) on an analog audio signal.

[c7] 7.The digital audio signal processing method of claim 6 wherein a sampling rate

of said pulse code modulation determines an interval between each sampling point in time domain.

[c8] 8.The digital audio signal processing method of claim 1 wherein said predetermined algorithm is a convolution algorithm.

[c9] 9.The digital audio signal processing method of claim 2 wherein said digital equalizer is a software program.

[c10] 10.The digital audio signal processing method of claim 1 wherein said second impulse response is generated from said first impulse response multiplied by a window function in time domain.